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IN REPLY REFER TO: Joint Interoperability Test Command (JTE) **6 Nov 08**

MEMORANDUM FOR DISTRIBUTION

SUBJECT: Extension of the Special Interoperability Test Certification of Nortel Defense Switched Network (DSN) Communications Server (CS) 1000M-Single Group (SG) and DSN CS1000M-Multi Group (MG) (including Voice over Internet Protocol [VoIP]) and DSN Meridian 1 (M1) Option 61C and DSN M1 Option 81C Digital Switching Systems with Software Release 4.5w and Product Enhancement Packages

References: (a) DoD Directive 4630.5, "Interoperability and Supportability of Information Technology (IT) and National Security Systems (NSS)," 5 May 2004
(b) CJCSI 6212.01D, "Interoperability and Supportability of Information Technology and National Security Systems," 8 March 2006
(c) through (g), see Enclosure

1. References (a) and (b) establish the Defense Information Systems Agency (DISA), Joint Interoperability Test Command (JITC), as the responsible organization for interoperability test certification.

2. The Nortel DSN CS1000M-SG Digital Switching System with Software Release 4.5w and Product Enhancements (including VoIP) is hereinafter referred to as the System Under Test (SUT). The SUT met all of its critical interoperability requirements and is certified as interoperable for joint use within the Defense Switched Network (DSN). The SUT is certified for VoIP specifically with certified Assured Services Voice Application Local Area Networks (ASVALANs) posted on the Unified Capabilities (UC) Approved Product List (APL). The JITC also determined, through analysis, that the Nortel DSN CS1000M-MG with VoIP, is also certified for joint use within the DSN. The analysis determined the DSN CS1000M-MG employs the same software and trunk/line card hardware as the Nortel DSN CS1000M-SG, and therefore is functionally identical to the Nortel DSN CS1000M-SG. The difference between the two switches is scalability. The DSN CS1000M-SG supports up to a maximum of 2000 ports and the DSN CS1000M-MG supports a maximum of 16,000 ports. When the SUT is fielded without VoIP, it is certified for Joint use within the DSN as well. The SUT without VoIP product is referred to and marketed within DoD as the Nortel DSN M1 Option 61C. Additionally, the DSN CS1000M-MG without VoIP is also certified for joint use within the DSN via the same analysis done on the CS1000M-MG with VoIP. The DSN CS1000M-MG without VoIP is referred to and marketed within DoD as the Nortel DSN M1 Option 81C. The listed test discrepancies shown in the SUT Interoperability Test Summary have an overall minor operational impact. One of the discrepancies, was with the Call Forwarding Variable (a conditional requirement), which does not properly interact with precedence calls above

ROUTINE and is therefore not certified for joint use within the DSN. The SUT was tested and met the critical interoperability requirements for the following DSN switch types: SMEO, Private Branch Exchange (PBX) 1, and PBX 2. No other configurations, features, or functions, except those cited within this report, are certified by the JITC, or authorized by the Program Management Office for use within the DSN. This certification expires upon changes that could affect interoperability, but no later than three years from the date of the original memorandum (14 September 2006).

3. The extension of this certification is based upon a desktop review. The original certification is based on interoperability testing conducted by JITC and a review of the vendor's Letters of Compliance (LoC). Testing was conducted at JITC's Global Information Grid Network Test Facility at Fort Huachuca, Arizona, from 21 November 2005 through 13 January 2006. Regression testing was conducted from 1 through 27 May 2006, and 19 through 30 June 2006. Patches were applied and additional regression testing was conducted 11 through 20 July 2006. Final regression testing was conducted 25 August 2006. Review of the LoC was completed on 30 July 2006 and documented in reference (c). Review of Product Enhancement Packages was completed on 7 September 2006. A desktop review was requested to include the following VoIP telephones: i2007 with firmware 0621C4J, i1140E with firmware 0625C4D, i1110 with firmware 0623C4D, and i1120E with firmware 0624C4D. These VoIP telephones were tested with the CS1000M-Single Group with Software Release 5.0w. The desktop review request was approved on 2 October 2008.

4. The interoperability test summary of the SUT is contained in Table 1. The SMEO required and conditional Capability Requirements (CRs) and Feature Requirements (FRs) are listed in Table 2. This interoperability test status is based on the SUT's ability to meet:

a. DSN services for Network and Applications specified in reference (d).

b. SMEO interface and signaling requirements for trunks/lines specified in reference (e) verified through JITC testing and/or vendor submission of LoC.

c. SMEO CRs/FRs specified in reference (e) verified through JITC testing and/or vendor submission of LoC.

d. Internet Protocol version 6 requirements specified in reference (e), paragraph 1.7, Table 1-4, by 30 June 2008 in accordance with reference (f) verified through vendor submission of LoC signed by the Vice President of the company.

e. The overall system interoperability performance derived from test procedures listed in reference (g).

Table 1. SUT Interoperability Test Summary

DSN Trunk Interfaces			
Interface & Signaling	**Critical**	**Status**	**Remarks**
T1 CAS (DTMF, DP)	Yes	Certified	Met all critical CRs and FRs with the following minor exceptions: The SUT does not restore the span to service within the required time duration.[1] The SUT recognizes a wink start signal greater than the specified maximum limit.[2] The SUT does not support glare hold resolution for their CAS trunks.[3]
T1 CAS (MFR1)	No	Not Tested	This interface is not supported.[4]
E1 CAS (DTMF, DP)	Yes (Europe only)	Certified	Met all critical CRs and FRs with the following minor exceptions: The SUT does not restore the span to service within the required time duration.[1] The SUT does not support glare hold resolution for their CAS trunks.[3] The on/off hook pulse that frames the preemption signal on the E1 CAS is intermittently out of the required tolerance of 100ms (+/-5ms).[5]
E1 CAS (MFR1)	No	Not Tested	This interface is not supported.[4]
T1 ISDN PRI NI 1/2 (ANSI T1.619a)	Yes	Certified	Met all critical CRs and FRs with the following minor exception: The SUT fails to automatically return trunks to a maintenance busy condition after the span is broken then restored.[6]
E1 PRI (ITU-T Q.955.3)	No (Europe only)	Certified	Met all critical CRs and FRs.
T1 SS7 (ANSI T1.619a)	No	Not Tested	This interface is not supported.[4]
E1 SS7 (ANSI T1.619a)	No	Not Tested	This interface is not supported.[4]
DSN Line Interfaces			
Interface & Signaling	**Critical**	**Status**	**Remarks**
2-Wire Analog (GR-506-CORE)	Yes	Certified	Met all critical CRs and FRs.
ISDN BRI NI 1/2	Yes	Certified	Met all critical CRs and FRs with the following minor exceptions: The SUT does not support NI2 BRI.[7] The only supported and certified interface is NI1 BRI with a single appearance of a single directory number.[8] The precedence above ROUTINE ringing cadence that the SUT applies to BRI phones does not meet the specifications.[9] The BRI instruments do not support precedence call waiting.[10]
2-Wire Proprietary Digital	No	Certified	Met all critical CRs and FRs.
VoIP (ITU-T H.323 Proprietary)	No	Certified	Met all critical CRs and FRs. Precedence call waiting indication is unique on VoIP phones.[11]
DSN Features and Capabilities			
Features and Capabilities	**Critical**	**Status**	**Remarks**
Common Features	No	Certified	Met all critical CRs and FRs with the following minor exceptions: The SUT does not correctly support the call forwarding variable feature.[12] The conference disconnect tone that is provided by the SUT does not meet the specifications.[13]
Attendant	No	Certified	Met all critical CRs and FRs with the following minor exceptions: Stations cannot be classmarked to prohibit the attendant console from performing a busy override to an active call.[14]
Public Safety	Yes	Certified	Met all critical CRs and FRs with the following exception: The SUT cannot perform a tandem call trace of a specified distant office directory number.[15]
Preset Conferencing	No	Not Tested	This feature is not supported.[16]

Table 1. SUT Interoperability Test Summary (continued)

DSN Features and Capabilities			
Features and Capabilities	**Critical**	**Status**	**Remarks**
Nailed-up Connections	No	Not Tested	This feature is not supported.[16]
PAT	No	Not Tested	This feature is not supported.[16]
DSN Hotline Services	Yes	Certified	Met all critical CRs and FRs with the following minor exceptions: The SUT does not support a protected hotline specified list.[17]
Network Management	Yes	Certified	Met all critical CRs and FRs.
Multiline Hunt Service	No	Certified	Met all critical CRs and FRs with the following minor exceptions: The SUT will not permit a BRI station to be a member of a multiline hunt group.[18]
ISDN Services (EKTS)	No	Not Tested	This feature is not supported.[16]
Synchronization	Yes	Certified	Met all critical CRs and FRs.
Reliability	Yes	Certified	Met all critical CRs and FRs.
Security	Yes	See note 19	See note 19
VoIP System	No	Certified	The SUT is certified for VoIP specifically with certified ASVALAN posted on the JITC TSSI program web page (http://jitc.fhu.disa.mil/tssi/cert_nortel) approved product list. See note 20.

Network Gateways				
Gateway	**Interface & Signaling**	**Critical**	**Status**	**Remarks**
PSTN	T1 CAS (DTMF, DP)	Yes	Certified	Met all critical CRs and FRs.
	E1 CAS (DTMF, DP)	No (Europe only)	Certified	Met all critical CRs and FRs.
	T1 ISDN PRI NI 1/2 (ANSI T1.607)	No	Certified	Met all critical CRs and FRs.
	E1 PRI (ITU-T Q.931)	No (Europe only)	Certified	Met all critical CRs and FRs.
	Ground Start Line	Yes	Certified	Met all critical CRs and FRs.
DRSN	TPC 2-Wire analog (GR-506-CORE)	Yes	Certified[21]	Met all critical CRs and FRs.

LEGEND:

ANSI - American National Standards Institute
ASVALAN - Assured Services Voice Application Local Area Network
BRI - Basic Rate Interface
CAS - Channel Associated Signaling
CRs - Capability Requirements
DISA - Defense Information Systems Agency
DoD - Department of Defense
DP - Dial Pulse
DRSN - Defense Red Switch Network
DSN - Defense Switched Network
DSS1 - Digital Subscriber Signaling 1
DTMF - Dual Tone Multi-Frequency
E1 - European Basic Multiplex Rate (2.048 Mbps)
EKTS - Electronic Key Telephone System
FRs - Feature Requirements
GR - Generic Requirement
GR-506 - LSSGR: Signaling for Analog Interfaces
GSCR - Generic Switching Center Requirements
H.323 - Standard for multi-media communications on packet-based networks
IPv4 - Internet Protocol version 4
IPv6 - Internet Protocol version 6
ISDN - Integrated Services Digital Network
ITU-T - International Telecommunication Union - Telecommunication Standardization Sector
JITC - Joint Interoperability Test Command
LSSGR - Local Access and Transport Area (LATA) Switching Systems Generic Requirements
Mbps - Megabits per second
MFR1 - Multifrequency Recommendation 1
MLPP - Multi-Level Precedence and Preemption
ms - millisecond
NI 1/2 - National ISDN 1 or 2
PAT - Precedence Access Threshold
PBX - Private Branch Exchange
PM - Program Manager
PRI - Primary Rate Interface
PSTN - Public Switched Telephone Network
Q.931 - Signaling Standard for ISDN
Q.955.3 - ISDN signaling standard for E1 MLPP
SMEO - Small End Office
SS7 - Signaling System 7
SUT - System Under Test
T1 - Digital Transmission Link Level 1 (1.544 Mbps)
T1.607 - ISDN – Layer 3 Signaling Specification for Circuit Switched Bearer Service for DSS1
T1.619a - SS7 and ISDN MLPP Signaling Standard for T1
TSSI - Telecom Switched Services Interoperability
TPC - Twisted Pair Copper
VoIP - Voice over Internet Protocol

Table 1. SUT Interoperability Test Summary (continued)

NOTES:

1 When any active trunk interface is physically broken and repaired, the SUT does not restore the span to service and remove the yellow alarm condition within the required time duration. In accordance with the GSCR paragraphs 7.1.4 and 7.2.2, the time required for the removal of the alarm condition after the physical restoration of a broken trunk is 15 (+/-5) seconds. The E1 CAS can take up to 90 seconds to restore, and all the other interfaces require 30 seconds to be restored. The operational impact is minor since the alarm clears automatically when the span is returned to service.

2 T1 CAS wink start signals greater than the specified maximum limit are recognized as valid by the SUT. The GSCR 5.3.3.3.1 and GSCR figure 3-2 defines the wink start recognition limits between 100 milliseconds (ms) to 350ms. The SUT recognizes wink start signals from 100ms to 925ms in duration. Since all certified switches within the DSN must generate the wink start signal within 140-290ms this anomaly has no operational impact.

3 The SUT does not support glare hold resolution on CAS trunks. It only supports glare release. Since the SUT is a subtending switch off of a Multifunction Switch (MFS) and all MFS support glare hold, complementing the SUTs capability to support glare release. Therefore, the operational impact is minor.

4 This interface is not supported. There is no operational impact because it is not a critical requirement.

5 The on/off hook pulse that initiates the preemption signal on the E1 CAS is intermittently out of the required tolerance of 100ms (+/-5ms). The pulse width was measured to be greater than 100 ms (highest @ 128 ms) about 20% of the time, but never had any impact on the ability of the SUT to support call preemption. Therefore, this anomaly has no operational impact.

6 If a T1 ISDN PRI interface is broken then restored when all channels are in a maintenance busy condition, the SUT fails to automatically return the channels to the previous busy condition. This anomaly has no operational impact because it only occurs when the SUT is in a maintenance condition.

7 The SUT does not support an National ISDN (NI)2 BRI interface. The only supported and certified BRI interface is NI1. The NI2 BRI interface is required for SMEO operation as specified by GSCR paragraph 2.3.3 Since the primary differences between NI1 and NI2 are supplemental features which currently are not fielded within the DSN nor are there plans to field them in the future, this anomaly has a minor operational impact.

8 The SUT will only support a BRI NI1 voice line with a single directory number and a single appearance of a directory number. However, multiple appearances with different directory numbers can be supported with the digital proprietary instruments which account for the majority of digital instruments fielded within the DSN. Therefore, the operational impact is minor.

9 The precedence above ROUTINE ringing cadence that the SUT applies to BRI phones does not meet the specifications as detailed in the GSCR 5.5.1. The precedence above ROUTINE cadence is distinct from the ROUTINE cadence when it is configured properly; therefore this anomaly has no operational impact.

10 The SUT does not support precedence call waiting for their BRI instruments; however the SUT does support precedence call waiting for all other phone types. Also, this requirement is conditional and therefore, has no operational impact.

11 The SUT supports the "call waiting" indication on VoIP telephones with visual indicators in lieu of audible tones as specified by the GSCR. When call waiting is invoked on a VoIP phone, the phone displays call waiting text along with a flashing symbol. The call waiting symbol flashes twice for a ROUTINE call and three times for precedence above ROUTINE call. Since the requirement for audible tone is conditional, and there are two visual indicators to alert the VoIP user of a waiting call, the operational impact of not supporting audible tones is minor.

12 When call forwarding variable (CFV) is assigned to any station on the SUT (except BRI; does not support CFV) and CFV is invoked by the user all precedence calls placed to that instrument are forwarded to the DSN or Public Switched Telephone Network (PSTN). Additionally any station with CFV invoked does not receive a “ping” ring when calls are being forwarded. Per the GSCR only ROUTINE precedence calls will be forwarded and precedence calls above are diverted to the attendant console, night service or alternate directory number, therefore this feature is not certified for use within the DSN. This feature is a conditional requirement and will have a minor operational impact.

13 The conference disconnect tone that is provided by the SUT does not meet the specifications designated in GSCR 5.5.2. The SUT conference disconnect tone is distinguishable from other DSN tones and cadences; therefore this anomaly has a minor operational impact.

14 Stations cannot be classmarked to prohibit the attendant console from performing a busy override to an active call, as specified in the GSCR 2.2.4. The proper override tone, however, is given to a station active with a call prior to the attendant’s bridging into the active call. Since attendants rarely bridge into calls and active calls remain connected when an attendant does bridge into a call, the operational impact is minor.

15 The SUT cannot perform a tandem call trace of a specified distant office directory number as specified in the GSCR. Since the SUT is predominately fielded within the DSN as a SMEO with no tandeming (e.g. subtending PBX1 or PBX2), this anomaly has a minor operational impact.

16 This feature is not supported. There is no operational impact because it is not a critical requirement.

17 The SUT will not allow the protection of a hotline call originator through the use of a hotline list as required by the GSCR. However, this capability can be accomplished with the SUT by classmarking authorized hotline users for receiving only calls from other hotline callers. The operational impact is minor.

18 The SUT will not permit an ISDN BRI station to be a member of a multiline hunt group. All other phone types can be configured as members of a multiline hunt group. Since ISDN BRI voice users are rarely used within the DSN and this feature can be accomplished on the SUT with analog and digital proprietary stations, this anomaly has a minor operational impact.

19 Security is tested by DISA-led Information Assurance test teams and published in a separate report.

20 An IPv6 capable system or product, as defined in the GSCR, paragraph 1.7, shall be capable of receiving, processing, and forwarding IPv6 packets and/or interfacing with other systems and protocols in a manner similar to that of IPv4. IPv6 capability is currently satisfied by a vendor Letter of Compliance signed by the Vice President of their respective company. The vendor must state, in writing, compliance to the following criteria by 30 June 2008**:**

a. Conformant with IPv6 standards profile contained in the DoD IT Standards Registry (DISR).
b. Maintaining interoperability in heterogeneous environments and with IPv4.
c. Commitment to upgrade as the IPv6 standard evolves.
d. Availability of contractor/vendor IPv6 technical support.

21 Interoperability Certification of the SUT does not constitute DRSN PM’s approval for connectivity to the DRSN. It is the user’s responsibility to request connectivity approval directly from the PM.

Table 2. SMEO Requirements

DSN Trunk Interfaces				
Interface	**Critical**	**Requirements Required or Conditional**		**References**
T1 SS7 (ANSI T1.619a) E1 SS7 (ITU-T Q.735.3) T1 CAS (MFR1) T1 CAS (DTMF, DP) E1 CAS (DTMF, DP) E1 CAS (MFR1) T1 ISDN PRI NI 1/2 (ANSI T1.619a) E1 ISDN PRI (ITU-T Q.955.3)	No No (Europe only) No Yes Yes (Europe only) No (Europe only) Yes No (Europe only)	Trunking	• Framing (R) • Line Code (R) • Signaling (R) • Alarms (R) • WWNDP (R) • Outpulsing digit formats (R: CAS only) • Routing (R) • Trunk Groups (R) • Call Processing (R) • CAS to CCS trunk interworking (C) • PCM-24/PCM-30 Interoperation (R) • Direct Inward Dialing (C)	• GSCR Sect. 7 • GSCR Sect. 7 • GSCR Sect. 5 • GSCR Sect. 2.5.7, 7.1.4 & 7.2.2 • GSCR Sect. 4.5.1 • GSCR Sect. 4.5.2 • GSCR Sect. 4.2 • GSCR Sect. 2.5.5 & 2.5.6 • GSCR Sect. 4 • GSCR Sect. 3.10 • GSCR Sect. 7.3 • GSCR Sect. 2.3.2
		Voice	• MOS (R) • MLPP (R) • Secure calls (R)	• CJCSI 6215.01B • GSCR Sect. 3 • CJCSI 6215.01B
		Facsimile	• Analog: TIA/EIA-465-A (R)	• DISR
		Data	• Modem (VBD) (R) • 56 kbps switched data (R: PRI only) • 64 kbps switched data (R: PRI only) • NX56 synchronous BER (R: PRI only) • NX64 synchronous BER (R: PRI only) • Secure data (STE/STU-III) (R)	• CJCSI 6215.01B • GSCR Sect. 3.10 • GSCR Sect. 3.10 • GSCR Sect. 3.10 • GSCR Sect. 3.10 • CJCSI 6215.01B
		VTC	• ITU-T H.320 (R: PRI only)	• DISR
DSN Line Interfaces				
Interface	**Critical**	**Requirements Required or Conditional**		**References**
2-Wire Analog ISDN BRI NI 1/2 (ANSI T1.619a) 2-Wire Proprietary Digital VoIP	Yes Yes No No	Access	• DN Identification (R) • Line Signaling (R) • Loop Start Line (R) • Alerting Signals and Tones (R) • WWNDP (R) • Call Processing (R) • Call Treatments (R) • 2W user access (R: 2-Wire Analog only) • Analog busy/idle (R: 2-Wire Analog only)	• GSCR Sect. 2.1.1 • GSCR Sect. 5.2 • GSCR Sect. 5.2.1 • GSCR Sect. 5.5 • GSCR Sect. 4.5 • GSCR Sect. 4.4 • GSCR Sect. 4.1 • GSCR Sect. 4.3.3 • GSCR Sect. 4.3.4.1
		Voice	• MOS (R) • Announcements (R) • MLPP (R) • Secure Calls (R)	• CJCSI 6215.01B • GSCR Sect. 3.1.3 • GSCR Sect. 3.4.3/3.9 • CJCSI 6215.01B
		Facsimile	• Analog: TIA/EIA-465-A (R)	• DISR
		Data	• Modem (VBD) (R) • 56 kbps switched data (R) • 64 kbps switched data (R: BRI only) • NX56 synchronous BER (R: BRI only) • NX64 synchronous BER (R: BRI only) • Secure data (STE/STU-III) (R)	• CJCSI 6215.01B • GSCR Sect. 3.10 • GSCR Sect. 3.10 • GSCR Sect. 3.10 • GSCR Sect. 3.10 • CJCSI 6215.01B
		VTC	• ITU-T H.320 (R: BRI only)	• DISR

Table 2. SMEO Requirements (continued)

DSN Features & Capabilities			
Feature/ Capability	**Critical**	**Requirements Required or Conditional**	**References**
Common Features	Yes	• Selective call rejection (C) • Denied originating service (C) • Code restriction and diversion (R) • Call waiting (C) • Three-way calling (C) • Add-on transfer and conference calling and call hold (C) • Call forwarding (C) • Call pick-up (C)	• GSCR Sect. 2.1.2 • GSCR Sect. 2.1.3 • GSCR Sect. 2.1.4 • GSCR Sect. 2.1.5 • GSCR Sect. 2.1.6 • GSCR Sect. 2.1.7 • GSCR Sect. 2.1.8 • GSCR Sect. 2.1.9
Attendant	No	• Initiate all precedence levels (C) • Visual display (C) • Override class of service (C) • Override busy line (C) • Call deflection (C) • Auto recall (C) • Waiting queue (C)	• GSCR Sect. 2.2.1 • GSCR Sect. 2.2.2 • GSCR Sect. 2.2.3 • GSCR Sect. 2.2.4 • GSCR Sect. 2.2.5 • GSCR Sect. 2.2.6 • GSCR Sect. 2.2.7
Public Safety	Yes	• E911 (C) • Trace of terminating calls (R) • Outgoing call trace (R) • Tandem call trace (R) • Trace of a call in progress (R)	• GSCR Sect. 2.4.1 • GSCR Sect. 2.4.2 • GSCR Sect. 2.4.3 • GSCR Sect. 2.4.4 • GSCR Sect. 2.4.5
Preset Conferencing	No	• Support 10 bridges; 1 originator and 20 conferees per bridge (C) • Assign up to 20 address numbers per bridge (C) • Use KXX codes for bridge access (C) • Conference notification recorded announcement (C) • Auto retrial and alternate address (C) • Bridge release (C) • Lost connection (C) • Secondary conferencing (C) • Address translation (C)	• GSCR Sect. 2.6 • GSCR Sect. 2.6 • GSCR Sect. 2.6 • GSCR Sect. 2.6.1 • GSCR Sect. 2.6.2 • GSCR Sect. 2.6.3 • GSCR Sect. 2.6.4 • GSCR Sect. 2.6.5 • GSCR Sect. 2.7
Nailed-up Connections	No	• Between any two like terminations (C) • PCM-24 and PCM-30, both CAS and CCS (C) • Supervision passed end-to-end for A/D or D/A (C) • Monitored and auto reconfigure (C) • Support at least 10% of circuits as nailed-up (C) • Non-preemptable (C)	• GSCR Sect. 2.8 • GSCR Sect. 2.8 • GSCR Sect. 2.8 • GSCR Sect. 2.8 • GSCR Sect. 2.8 • GSCR Sect. 2.8
PAT	No	• Classmark for/not for PAT screening (C) • 7 PAT mechanisms (C) • Outgoing call screening (C) • Functional structure (C) • Simultaneous calls limitation (C) • Overflow process (C) • Decrementing call-in-progress count (C) • Call treatment (C) • Queuing (C) • Attendant calls (C) • Operations measurement registers (C) • Maintenance and Administration of thresholds (C)	• GSCR Sect. 2.11.1 • GSCR Sect. 2.11.1 • GSCR Sect. 2.11.1.1 • GSCR Sect. 2.11.1.2 • GSCR Sect. 2.11.1.3 • GSCR Sect. 2.11.1.4 • GSCR Sect. 2.11.1.5 • GSCR Sect. 2.11.1.6 • GSCR Sect. 2.11.1.7 • GSCR Sect. 2.11.1.8 • GSCR Sect. 2.11.1.9 • GSCR Sect. 2.11.1.10

Table 2. SMEO Requirements (continued)

DSN Features & Capabilities			
Feature/ Capability	**Critical**	**Requirements Required or Conditional**	**References**
DSN Hotline Services	Yes	• Hotline restrictions (R) • Auto initiate (R) • Analog and digital (R) • Subscription basis (R) • Protected hotline calling (R) • WWNDP interoperable (R)	• GSCR Sect. 2.12 • GSCR Sect. 2.12 • GSCR Sect. 2.12 • GSCR Sect. 2.12 • GSCR Sect. 2.12.1-4 • GSCR Sect. 2.12.5
Network Management	Yes	• Interfaces (R) • Measurements and data generation (R) • Fault management (R) • Configuration management (R) • Accounting management (R) • Performance management (R) • NM controls (R) • Remote access (R)	• GSCR Sect. 9.1 • GSCR Sect. 9.2 • GSCR Sect. 9.3 • GSCR Sect. 9.4 • GSCR Sect. 9.5 • GSCR Sect. 9.6 • GSCR Sect. 9.7 • GSCR Sect. 9.8
ISDN Services	No	• EKTS (C)	• GSCR Sect. 10, table 10-3
Synchronization	Yes	• Line timing mode (R) • Internal Stratum 4 (R)	• GSCR Sect. 11.1.1.2 • GSCR Sect. 11.1.2.2
Reliability	Yes	• GR-512-CORE (R)	• GSCR Sect. 12
Security[1]	Yes	• DITSCAP (R)	• GSCR Sect. 13

VoIP			
Feature/ Capability	**Critical**	**Requirements Required or Conditional**	**References**
VoIP System	No	VoIP function is conditional. If VoIP is provided, **all** of the following requirements must be met: • Voice Quality with Mean Opinion Score of 4.0 or better • Class of Service (CoS) and Quality of Service (QoS) • ITU-T G.711 PCM Codec • Traffic Engineering • Security in accordance with DITSCAP • NM • Line timing • Internal Clock • Latency ≤ 60 ms • Packet Loss • IPv6 capable	• GSCR App. 3 • GSCR App. 3 • GSCR App. 3 • GSCR App. 3 • GSCR App. 3 • GSCR App. 3 • GSCR App. 3 • GSCR App. 3 • GSCR App. 3 • GSCR App. 3 • GSCR App. 3, paragraph 1.7

Network Gateways				
Gateway	**Critical**	**Requirements Required or Conditional**		**References**
PSTN[2]	Yes	Trunking	• Positive Identification Control (R) • On-Netting (R) • Off-Netting (R)	• CJCSI 6215.01B • CJCSI 6215.01B • CJCSI 6215.01B
DRSN[3]	Yes	Access	• Alerting Signals and Tones (R) • Call Processing (R) • Call Treatments (R) • Analog busy/idle (R)	• GSCR Sect. 5.5 • GSCR Sect. 4.4 • GSCR Sect. 4.1 • GSCR Sect. 4.3.4.1
		Voice	• MOS (C) • MLPP (C) • Secure calls (C)	• CJCSI 6215.01B • GSCR Sect. 3 • CJCSI 6215.01B

Table 2. SMEO Requirements (continued)

LEGEND:

2W	- 2-Wire	EIA	- Electronic Industries Alliance	PAT	- Precedence Access Threshold
A/D	- Analog to Digital Conversion	G.711	- Standard for PCM of Voice Frequencies	PCM	- Pulse Code Modulation
ANSI	- American National Standards Institute	GR	- Generic Requirement (Telcordia)	PCM-24	- Pulse Code Modulation - 24 Channels
App.	- Appendix	GSCR	- Generic Switching Center Requirements	PCM-30	- Pulse Code Modulation - 30 Channels
BER	- Bit Error Ratio	H.320	- Standard for Narrowband VTC	PRI	- Primary Rate Interface
BRI	- Basic Rate Interface	IEEE	- Institute of Electrical and Electronics Engineers, Inc.	PSTN	- Public Switched Telephone Network
C	- Conditional			Q.735.3	- SS7 Signaling Standard for E1 MLPP
CAS	- Channel Associated Signaling	IPv6	- Internet Protocol version 6	Q.955.3	- ISDN Signaling Standard for E1 MLPP
CCS	- Common Channel Signaling	ISDN	- Integrated Services Digital Network	R	- Required
CJCS I	- Chairman of the Joint Chiefs of Staff Instruction	IT	- Information Technology	Sect.	- Section
		ITU-T	- International Telecommunication Union - Telecommunication Standardization Sector	SMEO	- Small End Office
D/A	- Digital to Analog Conversion			SS7	- Signaling System 7
DISA	- Defense Information Systems Agency	LAN	- Local Area Network	STE	- Secure Terminal Equipment
DISR	- DoD IT Standards Registry	kbps	- kilobits per second	STU-III	- Secure Telephone Unit – 3rd Generation
DITSCAP	- DoD IT Security and Accreditation Process	KXX	- K= any number 2-8; X= any number 1-9	T1	- Digital Transmission Link Level 1 (1.544 Mbps)
DN	- Directory Number	Mbps	- Megabits per second	T1.619a	- SS7 and ISDN Signaling Standard for T1
DoD	- Department of Defense	MFR1	- Multi-Frequency Recommendation 1	TIA	- Telecommunications Industry Association
DP	- Dial Pulse	MLPP	- Multi-Level Precedence and Preemption	TIA/EIA-465A	- Group 3 Facsimile Apparatus for Document Transmission
DSN	- Defense Switched Network	MOS	- Mean Opinion Score		
DRSN	- Defense Red Switch Network	ms	- milliseconds	VBD	- Variable bit data
DTMF	- Dual Tone Multi-Frequency	NI 1/2	- National ISDN Standard 1or 2	VLAN	- Virtual LAN
E1	- European Basic Multiplex Rate (2.048 Mbps)	NM	- Network Management	VoIP	- Voice over Internet Protocol
		NX56	- Data format restricted to multiples of 56 kbps	VTC	- Video Teleconferencing
E911	- Emergency 911 Service	NX64	- Data format restricted to multiples of 64 kbps	WWNDP	- Worldwide Numbering and Dialing Plan
EKTS	- Electronic Key Telephone System				

NOTES:

1 Information assurance testing is accomplished via DISA-led Information Assurance test teams and published in a separate report.
2 Voice, facsimile, data, and VTC service requirements for PSTN are identical to DSN with the exception of MLPP.
3 Facsimile, data, and VTC services are not provided via the DSN to DRSN interface.

5. No detailed test report was developed in accordance with the Program Manager's request. JITC distributes interoperability information via the JITC Electronic Report Distribution (ERD) system, which uses Unclassified-But-Sensitive Internet Protocol Router Network (NIPRNet) e-mail. More comprehensive interoperability status information is available via the JITC System Tracking Program (STP). The STP is accessible by .mil/gov users on the NIPRNet at https://stp.fhu.disa.mil. Test reports, lessons learned, and related testing documents and references are on the JITC Joint Interoperability Tool (JIT) at http://jit.fhu.disa.mil (NIPRNet), or http://199.208.204.125 (SIPRNet). Information related to DSN testing is on the Telecom Switched Services Interoperability (TSSI) website at http://jitc.fhu.disa.mil/tssi.

6. The JITC point of contact is Captain Oskar Widecki, DSN 879-5269, commercial (520) 538-5269, FAX DSN 879-4347, or e-mail to Oskar.Widecki@disa.mil. The JITC's mailing address is P.O. Box 12798, Fort Huachuca, AZ 85670-2798. The tracking number for the SUT is 51824.

FOR THE COMMANDER:

for J. T. Schutte

Enclosure a/s

RICHARD A. MEADOR
Chief
Battlespace Communications Portfolio

JITC Memo, JTE, Extension of the Special Interoperability Test Certification of Nortel Defense Switched Network (DSN) Communications Server (CS)1000M-Single Group (SG) and DSN CS1000M-Multi Group (MG) (including Voice over Internet Protocol [VoIP]) and DSN Meridian 1 (M1) Option 61C and DSN M1 Option 81C Digital Switching Systems with Software Release 4.5w and Product Enhancement Packages

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ADDITIONAL REFERENCES

(c) Joint Interoperability Test Command, Memo, JTE, "Special Interoperability Test Certification of Nortel Defense Switched Network (DSN) Communications Server (CS) 1000M-Single Group (SG) and DSN CS1000M-Multi Group (MG) (including Voice over Internet Protocol [VoIP]) and DSN Meridian 1 (M1) Option 61C and DSN M1 Option 81C Digital Switching Systems with Software Release 4.5w and Product Enhancement Packages," 14 September 2006

(d) Chairman of the Joint Chiefs of Staff Instruction (CJCSI) 6215.01B, "Policy for Department of Defense Voice Services," 23 September 2001

(e) Defense Information Systems Agency, "Department of Defense Voice Networks Generic Switching Center Requirements (GSCR), Incorporated Change 1," 1 March 2005

(f) Executive Office of the President, "Transition Planning for Internet Protocol version 6 (IPv6)," 2 August 2005

(g) Joint Interoperability Test Command, "Defense Switched Network Generic Switch Test Plan (GSTP), Change 1, Revision 1," 1 June 2005